(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 202 948**
A2

(12) # EUROPEAN PATENT APPLICATION

(21) Application number: **86303918.6**

(51) Int. Cl.⁴: **G 11 B 20/00**

(22) Date of filing: **22.05.86**

(30) Priority: **22.05.85 JP 109640/85**
24.05.85 JP 111467/85

(43) Date of publication of application:
26.11.86 Bulletin 86/48

(84) Designated Contracting States:
DE FR GB IT NL

(71) Applicant: **CANON KABUSHIKI KAISHA**
30-2, 3-chome, Shimomaruko
Ohta-ku Tokyo(JP)

(72) Inventor: **Sakata, Tsuguhide**
956-3 Minami Ohya
Machida-shi Tokyo(JP)

(72) Inventor: **Kimura, Norio**
17-5 Ohsaki 1-chome Shinagawa-ku
Tokyo(JP)

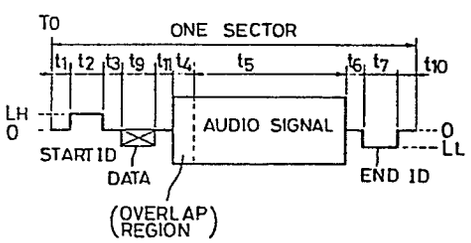
(72) Inventor: **Masahiro, Takei**
14-2 Chigusadai Midori-ku
Yokohama-shi Kanagawa-ken(JP)

(74) Representative: **Beresford, Keith Denis Lewis et al,**
BERESFORD & Co. 2-5 Warwick Court High Holborn
London WC1R 5DJ(GB)

(54) **Recording system and apparatus.**

(57) An information recording system and apparatus which records on a recording medium an audio signal, which is time-base compressed at a predetermined compression ratio, together with a date signal indicative of the compression ratio. The start signal indicative of the start of the audio signal, the data signal, the audio signal and the end signal indicative of the end of the audio signal are recorded in this order. A recording medium recorded by such system and apparatus is disclosed. A reproduction is also disclose which reproduces the information out of such recording medium.

FIG. 6



EP 0 202 948 A2

TITLE OF THE INVENTION

Information Recording System and Apparatus

BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates to information recording systems and apparatus, and more particularly to a recording system and apparatus which records an information signal, a start and an end signals representative of the start and end of the information signal, respectively, and a data signal related to the information signal on a recording medium.

Related Background Art

In recently proposed video floppy systems it is proposed to record a video signal as well as an audio signal on a magnetic sheet as a recording medium.

Fig. 1 diagramatically shows both the video and audio signals present together on a magnetic sheet. For example, according to the standards on video floppies, field video signals (V) and audio signals (A1, A2) for up to 50 tracks in all can be recorded on a magnetic sheet. The audio signal is converted by time base compression into the video band and recorded in FM modulated form. The recording time per track is approximately 10 seconds, assuming that the audio band is 5 KHz and the time base compression ratio is 640. Also, assume that the audio band is 2.5 KHz and the time

base compression ratio is 1,280, recording for approximately 10 seconds is possible. In this case, the audio signal may be completed in a single track or extended into the next track.

The ring-like audio track is divided into four sectors, as shown in Fig. 2. For example, in the where recording for 10 seconds is possible, an amount of information for 2.5 seconds is alloted to a single sector.

Fig. 3 illustrates the form of an audio signal and an additional signal to be recorded on a single sector.

In Fig. 3, after an elapse of t_1 from time T_0, a start ID signal having a level L_H continues for time t_2 for obtaining the timing to read audio signal information. After a blank of t_3 the audio signal starts. During the preceding interval of time t_4 the signal on a sector overlaps with that on the just preceding sector. This does not occur in the first sector corresponding to the leading portion of the audio signal. A new audio signal is in the remaining time t_5, the length of which is variable; 2.5 seconds at a maximum in the mode of 10 seconds per track. After a subsequent blank of t_6, an end ID signal having a level of L_2 continues for t_7 in order to detect the end of the audio signal and to obtain the timing of reading data

signal related to the subsequent audio signal. After a blank of t_8, the data interval continues for t_9. Then a blanking interval of time t_{10} appears, the length of which is varied according to the interval t_5 of the preceding audio signal, and thus one sector cycle ends. This form is basically the same in any of the sectors where the audio signal is present. The interval of the audio signal shown by t_5 is variable and selected so that the interval $t_5 + t_{10}$ is constant at all times. Thus the length of the audio signal determines the position of the end ID and data signals.

The data signal includes, in the form of digital data, the number indicative of the position, on the magnetic sheet, of a track on which the audio signal is recorded, the number of the leading and subsequent tracks through which audio signals are recorded continuously, the number of the tracks through which the video signal corresponding to the audio signal are recorded, the compression ratio of the audio signal, etc.

Fig. 4 shows an example of a device to reproduce an audio signal recorded in the form of a signal such as is mentioned above.

Reference numeral 1 denotes a magnetic sheet on which an audio signal, additional signals (ID and data signals) and a video signal such as are mentioned above are recorded. The magnetic sheet 1 is rotated at a constant speed by a motor 2 under control of a control

circuit including a microcomputer, etc., (hereinafter referred to as the "MICOM") 5. In this case, the rotation speed is 3,600 rpm according to the NTS system on the assumption that one complete rotation of sheet 1 causes a one-field video signal to be recorded.

A PG head 3 detects a PG mark (not shown) provided on sheet 1. A PG detection circuit 4 outputs a PG signal of one pulse per unit rotation of sheet 1 on the basis of the output of head 3. The output of the PG detection circuit 4 is applied to MICOM 5.

A magnetic reproduction head 6 picks up audio and additional signals and delivers them through an reproduction amplifier 7 to an FM demodulator 8 by which is produced a demodulated signal which is then deemphasized by a deemphasis circuit 9 and supplied to an A/D converter 10.

When MICOM 5 is given a reproduction command, it causes A/D converter 10 to perform sampling at a predetermined frequency f_{VC} and A/D conversion in accordance with the PG signal from PG detection circuit 4 and stores the output of the converter 10 in a memory 11.

As will be described later, the start point of the first sector of Fig. 2 is controlled at the time of recording so that it always has a constant phase relationship to the PG mark on sheet 1. Thus the sampling and A/D conversion by A/D converter 10 and

storage to memory 11 are performed according to the PG signal from the start point of the first sector.

In this way, when signals for one track are stored in memory 11, MICOM 5 starts to reproduce the audio signal while controlling an expansion ratio for audio reproduction, reproduction sequence, etc., on the basis of the data signal. That is, MICOM 5 reads the audio signals in memory 11 in address units and provides them to a D/A converter 12 at a frequency f_{AC} corresponding to the expansion ratio. The output of D/A converter 12 is produced as audible audio signals through a noise reduction circuit 13 from an output 14.

At this time, the position of head 6 on sheet 1 is controlled by a head drive circuit 15 under control of MICOM 5.

Fig. 5 is a flowchart for causing MICOM 5 to detect the start address for the audio signal in order to detect the data signal subsequent to the end ID signal, decode the contents of the data thereby to remove the additional signals, and extract the audio signal only.

The additional and audio signals reproduced by head 6 are stored in memory 11 (step 1). Initial conditions are set in counters, etc., required for detection of the additional and audio signals (step 2). The signals stored in memory 11 are read (step 3). It is determined whether the read signal is the start ID

signal represented by high (or low) level (step 4). If the read signal is not the start ID, control returns to step 3 and the next stored signal is read, which will be repeated until the start ID signal is detected. When the start ID signal is detected, a level counter in another area of memory 11 counts a high (or low) level of the ID signal (step 5). It is determined whether or not high (or low) level has been counted n times successively (step 6). The steps 3 - 6 are repeated until such state has been reached. This eliminates noise and ensures that the start ID signal will be detected. After the start ID signal has been detected, the level of the start ID signal is assigned to the bits in memory 11 corresponding to the number of the sector to which the start ID is applied (step 7).

X addresses are skipped in the addresses of memory 11 within the level of the start ID signal to the address of a point before the trailing edge of the start ID signal (step 8). Then, the trailing edge of the ID signal is detected, that is, the stored signals are again sequentially read out of memory 11 (step 9). It is determined whether or not the read data is a position signal having a level equal to 90 % of the high (or low) level of the ID signal and not a dropout (step 10). The steps 9 and 10 are repeated until the above conditions are satisfied. When the data signal is determined to be a position signal having a level equal to 90 % of the

high (or low) level of the start ID signal and to be not a dropout, the position signal is regarded as the trailing edge of the start ID signal, the start address of the audio signal is calculated from the address of the position signal, and this calculated address is stored in an address register set in another are of memory 11 (step 11).

In Fig. 3, while the start ID signal is shown sharply rising or falling, it will slowly rise or fall in fact. Therefore, only after the start ID signal has a value indicative of 90 % of its high (or low) level, and no dropout (a sharp level drop for only one - a few bits) is detected, the trailing edge of the start ID signal is detected. The start address of the location where the audio signal is stored is calculated by adding the address bits corresponding to the interval t_3 to the address of the trailing edge. Then the stored signals are read sequentially out of memory 11 at the start address of the audio signal calculated at step 11 (step 12). It is determined whether or not the end ID level is of high (or low) level (step 13). The steps 12 and 13 are repeated until the end ID is detected. After the end ID has been detected and in order to ascertain the end ID, the above level counter counts high (or low) levels (step 14). It is determined whether or not n successive high (or low) levels have been counted (step 15). The steps 12 - 16 are repeated

until n successive high (or low) levels are counted. When n high (or low) levels have appeard, address skip is performed by z addresses to the address of a point immediately after the trailing edge of the end ID (step 16). After this skip, a synchronizing signal (hereinafter referred to as the "sync" contained in the data signals is detected out of the signals stored in memory 11 (step 17). It is determined whether or not the sync has been detected (step 18). The steps 17 and 18 are repeated until the sync has been detected. After the sync has been detected, the data signal is read (step 19). The contents of the data signal are decoded by MICOM 5 (step 20). The result is stored in a data signal register provided in another area of memory 11 (step 21). Then sectors are then counted by a sector counter provided in another area of memory 11 (step 22). It is determined whether or not the above flow operation has been performed on each of the four sectors by determining whether the contents of the sector counter is 4 (step 23). When the flow operation has not been completed for all the four sectors, address skip or jump is performed by y addresses to the address of a point immediately before the leading edge of the start ID of the next sector (step 24). The steps 3 - 23 are repeated for all the four sectors. Correct data information is determined by the information in all the four sectors and stored in a register provided in another

area of memory 11.

In the above manner, the audio start address of each sector and correct information are obtained. However, if the data signal follows the end ID, as mentioned above, the detection of the data signal must be performed after detection of the start and end IDs. That is, detection of the end ID is indispensable. In addition, since the length of the audio signal between the start ID and the end ID is variable, jump in address from the start ID to the end ID is impossible, the signals stored in memory 11 must be retrieved through the length of the audio signal until the end ID is detected. According to the above flow, the steps 12 - 16 correspond to the processing from the detection of the start ID to the detection of the end ID. If the audio signal has been recorded at its maximum length in a sector, it will take a few seconds for MICOM 5 to detect the end ID, which is a large time loss compared to the case where the steps 12 to 16 are omitted. On the other hand, when the video and audio signals are reproduced concurrently, the information obtained by the above flow is used for control of the reproduction. Therefore the audio reproduction starts after a time loss of a few minutes and hence great non-coincidence in time between the reproduced video and audio signals results. In video-audio concurrent reproduction, when the data signal is being reproduced while being reproduced, no

information on the data signal is available until after the end ID and hence the information on the compression ratio of the audio signal as mentioned above is not available. This renders concurrent reproduction of video and audio signals substantially impossible.

As described above, since the data signal is after the end ID, the access to data is slow, which, as a result, will be an obstacle to the reproduction processing.

In addition, if, for example, two memories, each memory having a memory capacity for one sector, should each be provided for a respective one of an even and an odd sectors and the stored signals should be read out of the corresponding memory while being recorded in the corresponding sectors, the tail end portion of each sector, i.e. the data signal portion, may be damagingly superimposed on the signal in the next sector. In this case, the reproduction processing of these data signals would be substantially impossible.

SUMMARY OF THE INVENTION

An object of the present invention is to provide an information recording system which records an information signal together with the start and end signals indicative of the start and end of the information signal and the related data signal so that in reproduction, the system has a very short access time

for the data signals.

Another object of the present invention is to provide an information recording system which records an information signal together with the start and end signals indicative of the start and end of the information signal and the related data signal so that data signals are hardly damaged.

According to a preferred embodiment of the present invention for attaining the above objects, there is disclosed an information recording apparatus which records a start signal indicative of the information signal, a data signal on the information, the information signal, and an end signal indicative of the end of the information signal in this order.

Still another object of the present invention is to provide an information recording system which records an information signal together with the start and end signals indicative of the start and end of the information signal and the related data signal so that the respective signals except the information signal are combined at correct timings.

Other objects of the present invention will be apparent from the following detailed description of embodiments thereof.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 shows recording tracks on a magnetic

sheet;

Fig. 2 shows an audio track consisting of sectors;

Fig. 3 shows a conventionally proposed arrangement of an audio signal, an ID signal and a data signal;

Fig. 4 is a block diagram of one example of a reproduction apparatus which is capable of reproducing the signals of Fig. 3;

Figs. 5A, 5B and 5C are a flowchart showing a portion of the operation flow for MICOM of Fig. 4 when the signals in the arrangement of Fig. 3 are reproduced.

Fig. 6 shows one example of signal arrangement of the present invention applied to the audio recording described in Figs. 1 - 3;

Figs. 7A and 7B and 8A and 8B are flowcharts showing a portion of the operation flow for MICOM of Fig. 4 for reproducing the signals arranged in accordance with the arrangement of Fig. 6;

Fig. 9 is a block diagram of a recording apparatus as one embodiment of the present invention for recording the signals following the arrangement of Fig. 6; and

Fig. 10 illustrates allocation of the areas of the memory 11 in which the signals are stored.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Now one embodiment of the present invention in

which the present invention is applied to the audio recording described in Figs. 1 - 5 will be described.

Fig. 6 shows the form of the audio and additional signals according to the present invention recorded in one of the sectors mentioned above. After an elapse of t_1 from T_0, start ID signal to obtain the timing of reading the audio signal continues for t_2. After a blank of t_3, the data signal continues for t_9. Then after a blank time t_{11}, the audio signal beings. For the preceding time t_4, the signal on this sector overlaps that on the preceding sector. However, there is no overlapping portion either in the first sector of a single track within which the whole audio signal is completed or in the first sector of the starting one of the tracks through which the audio signal are continuously recorded. The remaining sector portion for t_5 is for a new audio signal and the length of this sector portion is variable. Subsequently, after a blank of t_6, the end ID signal to detect the end of the audio signal continues for t_7. Then after a blank t_{10}, the next sector appears.

Fig. 7 is the preferable operation flow of MICOM 5 for reproduction of the signals, recorded in the arrangement of Fig. 6, using the apparatus of Fig. 4.

In Fig. 7, a block 100 corresponds to steps 1 to 11 of Fig. 5; a block 101 corresponds to the steps 17 to 25 of Fig. 5. In block 100, in the same way as in

the steps 1 to 11 of Fig. 5, the start ID signal is detected out of the signals stored in memory 11. After ascertainment of the ID signal, the sector number and level are assigned to the start ID signal and the start address of the audio signal is calculated and stored. Immediately thereafter, in block 101, in the same way as in steps 17 to 25 of Fig. 5, the data signal is detected and decoded, and the contents of a correct data signal are stored by the data signals for the four sectors.

Fig. 8 shows a flow for audio reproduction on the basis of the information obtained in accordance with the flow of Fig. 7. First, initial conditions such as the expansion ratio, i.e., the frequency for reading data out of memory 11, etc., are set using the information derived in accordance with the information of Fig.7 (step 26). The address read out of memory 11 is set in the audio start address derived in accordance with the flow of Fig. 7 (step 27). On the other hand, the level of the start ID signal assigned previously in the flow step 7 of Fig. 7 is loaded (step 28). It is then determined whether or not there is an empty sector (step 29). In this case, when the start ID signal is of low level, it is predetermined that the sector to which the start ID signal has been annexed is an empty sector in which no audio signals are recorded. Thus the level assignment of start ID signal is loaded at step 23, i.e.

the level of the start ID signal is detected for being high or low. If this level is high, control passes to step 30. If low, control branches to END.

When it is detected that the level of the start ID signal assigned in step 29 is high and the sector is not empty, signals are read out of memory 11 at the audio signal start address (step 30) and the end ID signal is detected (step 31). When the end ID signal is not detected, the flow goes to step 36. When the end ID signal is detected the flow branches to step 32.

Now, the process performed when the end ID signal has not been detected will be described.

At step 31, when it is not determined that the data is not the end ID signal, the difference counter is incremented (step 36) and it is determined whether or not the count of the difference counter represents a delay corresponding to t counts (step 37). The steps 31 to 37 are repeated to retrieve memory 11 until the count of the difference counter reaches t counts. When the count has reached t, the level counter is cleared, and the address consisting of the audio start address plus t address is stored in a register at an address 2 provided in another area of memory 11. On the other hand, the address delayed t addresses relative to the address stored in the register at an address 2, i.e. the start address of the audio signal, is stored in the register at an address 1 (step 38). The signal is read

out of memory 11 using the address in address of 2 (step 39). It is determined whether or not the read data is the end ID signal (step 40). If it is not the end ID signal, the signal is read out of memory 11 using the address in address 1 (step 41). The read signal is output as an audio signal to D/A converter 12 (step 42). The addresses in addresses 1 and 2 are then incremented (step 43). Control then returns to step 39. Then the detection of the end ID signal continues while audio signal is being reproduced. When the end ID has been detected and in order to ascertain that the data is the end ID signal, the level counter is incremented (step 44). It is determined whether or not the level counter has counted up to n (step 45). If the count has not reached n, the data cannot be regarded as the end ID and the data is read out of memory 11 using the address in address 1 (step 46). The read data is output to D/A converter 12 (step 47). The addresses in addresses 1 and 2 are incremented (step 48). Then control returns to step 39 and audio signals continue to be output until the end ID is ascertained. When the read data is ascertained as the end ID, the sector counter is incremented (step 49). Then it is determined whether or not the processing has reached sector 4 (step 50). When the processing has not reached, it passes to the next sector and repeats the above flow. On the other hand, when the processing has reached sector 4, the

processing ends.

As described above, in the particular embodiment, arrangement is such that the end ID may be read in advance in accordance with the address in address 2 which has been incremented by t addresses over the address in address 1 at which the audio signal is read.

When the end ID signal has been detected in accordance with the address in address 2 while the audio signal in a sector is being read in accordance with the address in address 1, the flow passes from step 40 to steps 44, 45. Then it is determined whether or not the count of the sector counter is 4, i.e. whether or not all the audio signals in the four sectors provided on a single track have been read. If all the audio signals have not been read, the flow branches from step 50 to step 27 to set the address in address 1 used for reading the audio signal at the audio start address in the next sector. The steps 30 and subsequent steps mentioned above are then executed to reproduce the audio signal in the next sector.

When, at step 31, the end ID has been detected and the flow has branched to step 31, the level counter is incremented for ascertaining the end ID counts (step 32). The difference counter is incremented to reproduce the audio signal with a delay of addresses corresponding to a time duration of t from the address at which the

level counter is detecting the end ID (step 33). It is determined whether or not the level counter has counted n times (step 34). If the answer is no, the step 30 and the subsequent steps will be repeated. If the level counter has counted n times, it is determined whether the difference counter has counted up to t (step 35). If the answer is no, the step 30 and the subsequent steps will be repeated. If the answer is yes, the audio signal is regarded as being absent at that time and the operation ends.

As described above, according to the signal arrangement of Fig. 6, the reproduction access time for the data signal stored in memory 11 is extremely shortened. Thus the audio signal can be very well reproduced on the basis of the data signal.

If two memories each having a capacity for one sector are assigned, one for each of an add and an even sectors and the signals stored in the memories are read alternately in sector units and recorded on sheet 1, the data signal is recorded before the end ID which is hardly prone to be superimposed although there may be a slight change in the timing of switching the memory. Thus the data is recorded successfully without being damaged by superimposition.

The above embodiment first derives information from the start ID and the data signal of the recorded signal including the start ID, data signal, audio signal

and end ID signal and reproduces the audio signal. If the steps of detecting the start ID and decording the data signal are provided in the setting of initial conditions such as the expansion ratio of the audio data in step 26 of the flowchart of Fig. 8, audio reproduction can be performed on a real time basis without performing the flow operation of Fig. 7.

Namely, in the flowchart of Fig. 7, all the data signals in the four sectors are detected and a correct data signal is determined from these four data signals, as mentioned above. Therefore, when the expansion ratio of the audio data, etc., are initially set, a time lag for the time required for such determination will occur; i.e. the time lag will occur when control passes from step 26 to step 27. However, if, after the start ID of a sector has been detected, the expansion ratio of the audio data, etc., are initially set on the basis of the data signal of that sector, the time lag can be shortened and audio reproduction can be performed on an improved real time basis.

A recording apparatus suitable for forming and recording the signal having the arrangement shown in Fig. 3 or Fig. 6 will now be described with respect to Fig. 9.

Fig. 9 is a block diagram of such recording apparatus. In Fig.9, the same reference numeral as in Fig. 4 denotes the same element as that described above.

In Fig.9, the audio signal is input to an terminal 21, passes through a noise reduction circuit 22 to an A/D converter 23 where the audio signal is sampled at a sample frequency f_{AS} from a controller including a microcomputer (hereinafter referred to as the "MICOM") 5, etc., and converted into a digital signal. The digital audio signal for one sample is sent to MICOM 5. MICOM 5 stores this signal is internal memory and then transfers this signal to memory 11 for writing.

Assume that memory 11 has a capacity for one track, for simplification of description. The digital audio signal is stored in memory 11, as shown in Fig. 10. In Fig. 10, the start address 100 of memory 11 has a given phase relationship to the PG mark (not shown) on magnetic sheet 1 when the timing generator (not shown) within MICOM 5 is reset by the output signal from PG detection circuit 4. The end address 117 of memory 11 represents the memory address immediately before the said resetting is performed. Areas 101 - 116 provide a memory area corresponding to one track. In Fig. 10, the ditized audio signal is written by MICOM 5 into memory areas 102, 106, 110 and 114 excluding predetermined memory areas, i.e. shown by reference numerals 101, 103 - 105, 107 - 109, 111 - 113, 115 and 116 of Fig. 10.

When the writing of the audio signal has been completed, MICOM 5 writes the start ID signal, end ID

signal, and data signal in accordance with the predetermined signal format mentioned above into the predetermined areas of memory 11, i.e. shown by reference numerals 101, 103, 105, 107 - 109, 111 - 113, 115 and 116. That is, if the format of Fig. 3 is obeyed, the start ID signals are written into the areas 101, 105, 109, 113; the end ID signals written into the areas 103, 107, 111, 115; and the data signals written into areas 104, 108, 112, 116. Note that if the format shown in Fig. 6 is obeyed, the above areas will be slightly modified.

When the respective signals are written by MICOM 5 in the arrangement obeying the predetermined format into memory 11 in order of address in the memory. MICOM 5 produces a recording gate signal corresponding to one round of sheet 1 on the basis of the PG signal from PG detector 4. The gate signal opens the gate for recording amplifier 28. This causes a recording current for one round to flow through head 29 and recording to be performed on one track. In this case, the contents of memory 11 are continuously read at a read rate f_{VC} in synchronism with PG signal. Thus D/A converter 25 is supplied continuously with signals for four sectors, one for each such format and produces an output containing an analog signal waveform corresponding to its digital input signal. The output of D/A converter 25 is delivered through preemphasis circuit 26 to FM modulator

27 where, after FM modulation, the signal is delivered to recording amplifier 28. The FM signal is present at all times in synchronism with the PG signal. The gate of recording amplifier 28 is opened for an interval of one round in synchronism with the PG signal for recording of the one-track signal.

The sheet rotating motor 2 rotates sheet 1 under control of MICOM 5, for example, at 3,600 rpm in the case of NTSC system. The positional control of head 29 on magnetic sheet 1 is performed by head driver circuit 15 under control of MICOM 5.

As described above, in the particular embodiment, when the audio signal is compressed, and recorded together with the ID signals and the data signal, the audio signal is first sampled and A/D converted, and the resulting signal data are stored in sequence in memory 11 at addresses excluding the addresses corresponding to the positions where the ID signal and the data signal are to be inserted in accordance with a predetermined format. After the storage of the audio signal has been completed, the ID signals and the data signals are now stored in the addresses, which remained empty during storage of the audio signal, such that the resulting signal arrangement in memory 11 obeys the predetermined format. Thus if the information stored in memory 11 is then read in sequence in order of address, the signals arranged in accordance with the predetermined format

will be obtained. This method is very useful when it is necessary to read the contents of memory 11 at very high speed, as when it is required to compress and record information at very high speed. That is, since it is only required to read the contents of the memory simply in order of address, no complexed hardware is needed, very high reading of the signals is possible and the compression ratio can be increased greatly. On the other hand, when the memory is written, there is a relatively much time margin. Therefore, it is very rational to perform an operation such as that mentioned above during writing, and hardware load is reduced.

In a modification, the writing of signals into memory 11 may be performed in the order of signals arranged in accordance with a predetermined format because there is a relatively much time margin during writing. On the contrary, first, the ID signals and the data signal may be stored in their respective addresses and the audio signal may then be stored in the remaining address.

In the particular embodiment, the memory 11 has been described as having a capacity for a track, i.e. four sectors. However, the frequency f_{VC} may be set such that if the memory has a capacity for one sector, it will take 1/240 seconds for all the information in the memory to be read, and that if the memory has a capacity for one track, it will take 1/60 seconds for all

the information to be read (this implies that sheet 1 is rotated at 3,600 rpm).

According to the present embodiments, the start ID signal, data signal and end ID signal are stored in memory areas excluding the area where the audio signal is stored. Thus, advantageously, complex timing control can be avoided, both the signals can be easily and well adjusted in timing, and hardware load and hence cost can be reduced compared with the method including the steps of storing only an audio signal in a memory, and reading the audio signal out of the memory while causing a microcomputer to produce the start ID signal, data ID signal, and end ID signal at appropriate timing.

While the above embodiment exemplifies audio recording to which the present invention is applied, of course, the invention disclosed in the particular embodiment is not limited to this.

As described above, according to the particular embodiment, an information signal is recorded together with the start signal, end signal, and the related data signal such that the access time for the data signal during reproduction is extremely shortened and that the data signal is well recorded without being damaged by other signals. This is very advantageous for information recording and reproducing apparatus of this type.

CLAIMS:-

1. An information recording apparatus comprising:

a) means for supplying a start signal, a data signal, an information signal and an end signal indicative of the end of said information signal; and

b) means for writing into a medium said start signal, said data signal, said information signal and said end signal, supplied by said supplying means in this order.

2. An information recording apparatus according to claim 1, wherein said information signal is an audio signal.

3. An information recording apparatus according to claim 2, wherein said audio signal is a signal which is compressed in time base.

4. An information recording apparatus according to claim 3, wherein said supplying means includes:

a) means for producing a pair of said start signal and said data signal; and

b) means for combining said information signal with said data signal and said data signal.

5. An information recording apparatus according to claim 1, wherein said writing means includes means for

modulating said start signal, said data signal, said information signal and said end signal and writing these signals in this order on a medium.

6. An information recording apparatus according to claim 3, wherein said data signal represents the time-base compression ratio for said audio signal.

7. An apparatus for recording an information signal, comprising:

a) means for producing a pair of a start signal and an end signal to sandwich the information signal therebetween;

b) means for providing a data signal related to said information signal;

c) means for combining said start signal, said data signal, said information signal, and said end signal in the named order; and

d) means for recording the combined signal produced form said combining means.

8. An apparatus according to claim 7, wherein said information signal is an audio signal.

9. An apparatus according to claim 8, wherein said information signal is an audio signal compressed in time base.

10. An apparatus according to claim 9, wherein said data signal is a signal indicative of the time base compression ratio for said audio signal.

11. An apparatus according to claim 7, wherein said recording means includes means for modulating said pair of a start signal and an end signal, and said information signal.

12. An apparatus according to claim 7, wherein said combining means includes:

a) memory means for storing said information signal at a predetermined part of said memory means;

b) control means for storing said pair of a start signal and an end signal, one in each of the parts of said memory means adjacent to both sides of said predetermined part of said memory means where said information signal is stored.

13. An apparatus according to claim 12, wherein said recording means includes:

means for reading said start signal out of said memory part of said memory means where said start signal is stored.

14. An information recording method for recording on a medium an audio signal compressed in time base at

a predetermined compression ratio, and a data signal indicative of the compression ratio, said method comprising the steps of:

a) a first step of recording on said medium a start signal indicative of the start of said audio signal;

b) a second step of recording said data signal on said medium subsequent to said first step; and

c) a third step of recording said audio signal on said medium subsequent to said second step.

15. An information recording method according to claim 14, further including:

d) a fourth step of recording an end signal indicative of the end of said audio signal.

16. An information recording method according to claim 14, wherein said first and second steps include FM modulation and recording.

17. An information recording method according to claim 14, wherein said medium is of a disc-type.

18. An information recording method according to claim 14, wherein said medium includes a magnetic recording medium.

19. A method of recording an information signal,

comprising the steps of:

a) producing a pair of a start signal and an end signal to sandwich said information signal therebetween;

b) producing a data signal related to said information signal;

c) combining said start signal, said data signal, said information and the end signal in the named order; and

d) recording the combined signal obtained through the combining step.

20. A method according to claim 19, wherein said information signal is an audio signal.

21. A method according to claim 20, wherein said audio signal is an audio signal compressed in time base.

22. A method according to claim 21, wherein said data signal is a signal indicative of the time base compression ratio for said audio signal.

23. A method according to claim 19, wherein said pair of a start signal and an end signal is indicative of the status of said information.

24. An information recording method for recording on a medium an information signal together with a data

signal related to the information signal, comprising the steps of:

a) a first step of recording on said medium a start signal indicative of the start of said information signal on said medium;

b) a second step of recording said data signal on said medium subsequent to said first step; and

c) a third step of recording said information on said medium subsequent to said second step.

25. An information recording method according to claim 24, further comprising:

d) a fourth step of recording on said medium an end signal indicative of the end of said information signal subsequent to said third step.

26. An information recording method according to claim 24, wherein said information signal is an audio signal.

27. An information recording method according to claim 26, wherein said information signal includes an audio signal compressed in time base at a predetermined compression ratio.

28. An information recording method according to claim 27, wherein said data signal is a signal indicative

of said compression ratio.

29. An information recording method according to claim 24, wherein said medium is a disk-like recording medium.

30. An information recording method according to claim 24, wherein said medium includes a magnetic recording medium.

31. A reproduction apparatus for reproducing an audio signal out of a medium in which a start signal indicative of the start of said audio signal compressed in time base at a predetermined ratio, a data signal indicative of said compression ratio, said audio signal and an end signal indicative of the end of said audio signal are recorded in this order, comprising:

a) means for detecting the position on said medium where said start signal is recorded; and

b) means for reading said data signal out of the detection result by said detecting means.

32. A reproduction apparatus according to claim 31, wherein said detecting means includes means for detecting the position where said start signal rises or falls.

33. A reproduction apparatus according to claim 32, wherein said reading means includes means for detecting and reading said data signal in accordance with the position where said start signal rises or falls.

34. A reproduction apparatus according to claim 33, further comprising:

means for expanding in time base said audio signal, compressed in time base, in accordance with the data signal read by said reading means.

35. A recording medium on which an audio signal compressed in time base at a predetermined compression ratio is recorded in a block indicated by a start signal and an end signal, wherein a signal indicative of said compression ratio is recorded between said start signal and said audio signal.

36. A recording medium according to claim 35, wherein said recording medium includes a disk-like medium.

37. A recording medium according to claim 35, wherein said medium includes a magnetic recording medium.

38. A recording medium according to claim 36, wherein said disk-like recording medium inlcudes a

plurality of concentric tracks formed thereon, each track including four said blocks.

FIG. 1

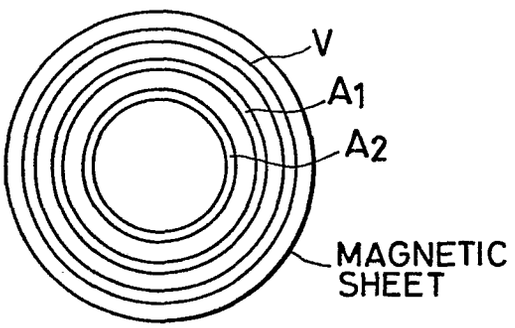


FIG. 2

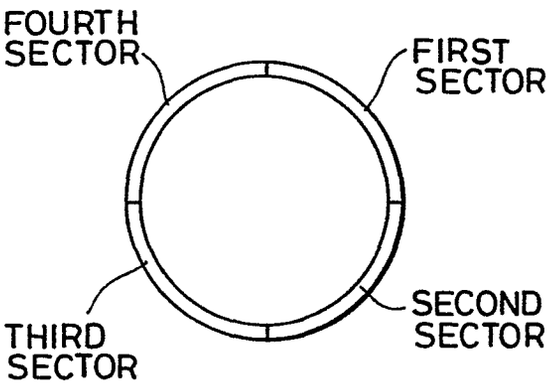


FIG. 3

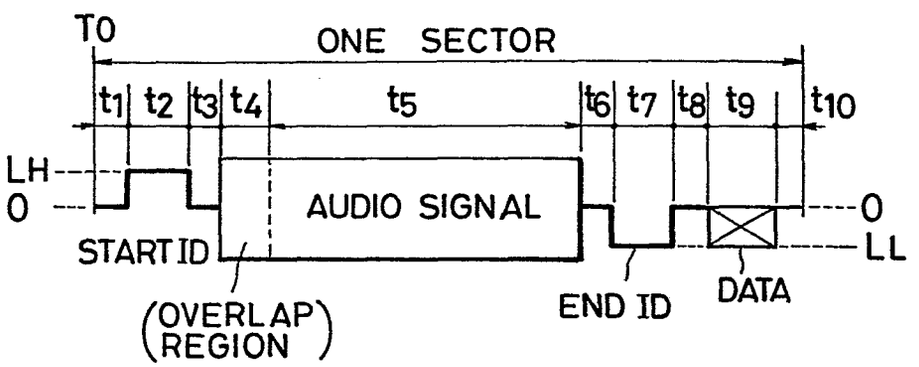


FIG. 4

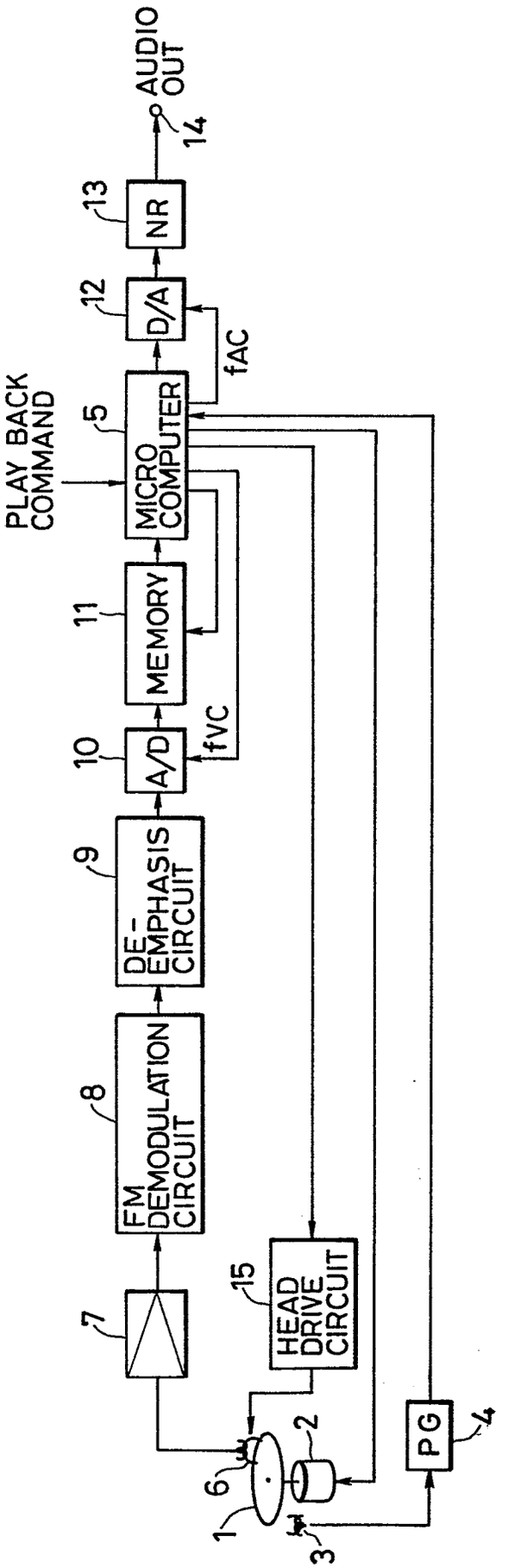

AUDIO OUT
14
13
NR
12
D/A
fAC
PLAY BACK COMMAND
5
MICRO COMPUTER
11
MEMORY
10
A/D
fVC
9
DE-EMPHASIS CIRCUIT
8
FM DEMODULATION CIRCUIT
7
15
HEAD DRIVE CIRCUIT
6
1
2
3
PG
4

FIG. 5A

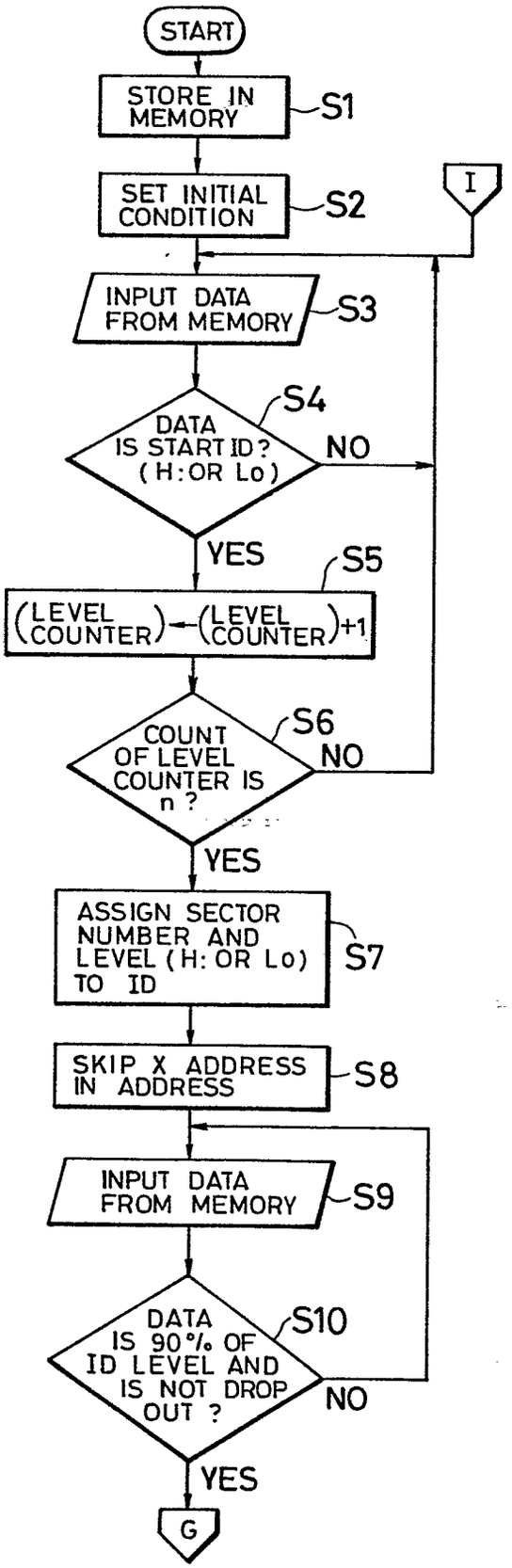

START
STORE IN MEMORY
S1
SET INITIAL CONDITION
S2
I
INPUT DATA FROM MEMORY
S3
DATA IS START ID? (H:OR Lo)
S4
NO
YES
(LEVEL COUNTER) ← (LEVEL COUNTER)+1
S5
COUNT OF LEVEL COUNTER IS n ?
S6
NO
YES
ASSIGN SECTOR NUMBER AND LEVEL (H: OR Lo) TO ID
S7
SKIP X ADDRESS IN ADDRESS
S8
INPUT DATA FROM MEMORY
S9
DATA IS 90% OF ID LEVEL AND IS NOT DROP OUT ?
S10
NO
YES
G

FIG. 5B

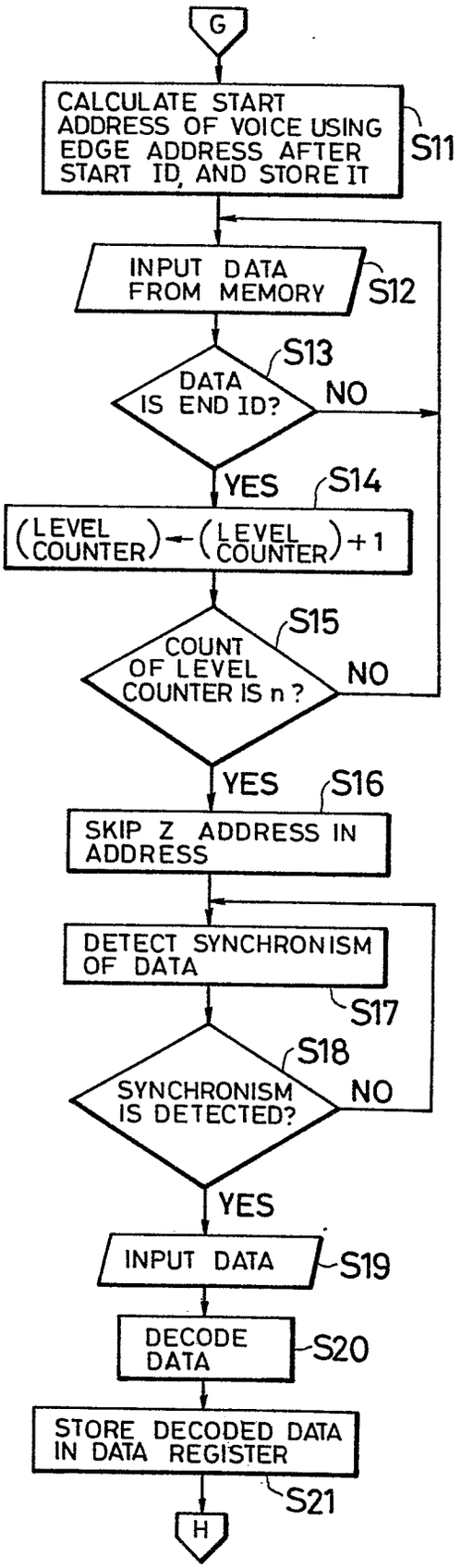

G
CALCULATE START ADDRESS OF VOICE USING EDGE ADDRESS AFTER START ID, AND STORE IT
S11
INPUT DATA FROM MEMORY
S12
DATA IS END ID?
S13
NO
YES
(LEVEL COUNTER) ← (LEVEL COUNTER)+1
S14
COUNT OF LEVEL COUNTER IS n ?
S15
NO
YES
SKIP Z ADDRESS IN ADDRESS
S16
DETECT SYNCHRONISM OF DATA
S17
SYNCHRONISM IS DETECTED?
S18
NO
YES
INPUT DATA
S19
DECODE DATA
S20
STORE DECODED DATA IN DATA REGISTER
S21
H

FIG. 5C

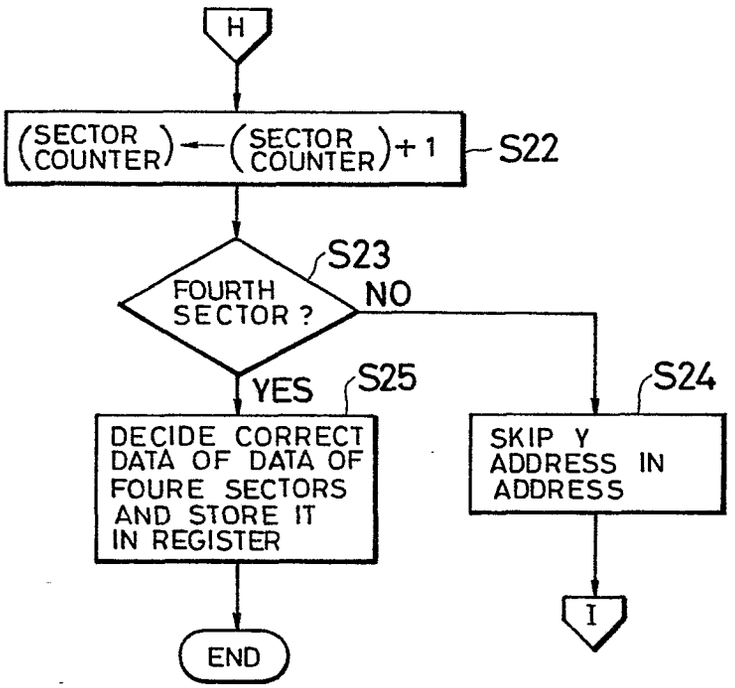

H
(SECTOR COUNTER) ← (SECTOR COUNTER) + 1
S22
FOURTH SECTOR ?
S23
NO
YES
S25
S24
DECIDE CORRECT DATA OF DATA OF FOURE SECTORS AND STORE IT IN REGISTER
SKIP Y ADDRESS IN ADDRESS
I
END

FIG. 6

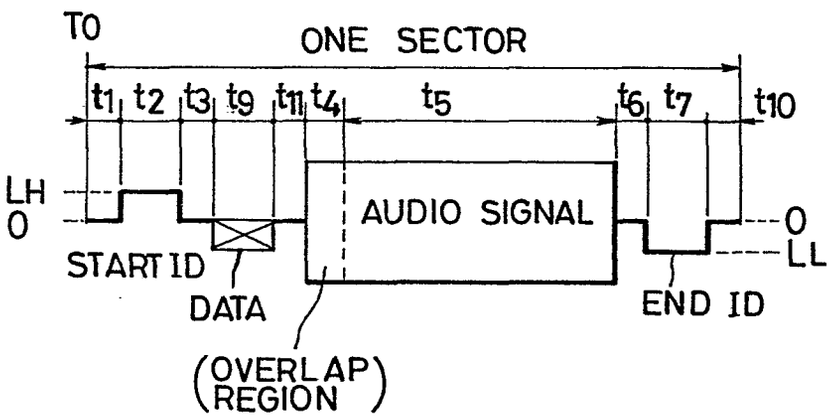

T0
ONE SECTOR
t1 t2 t3 t9 t11 t4 t5 t6 t7 t10
LH
0
AUDIO SIGNAL
0
LL
START ID
DATA
END ID
(OVERLAP REGION)

FIG. 7A

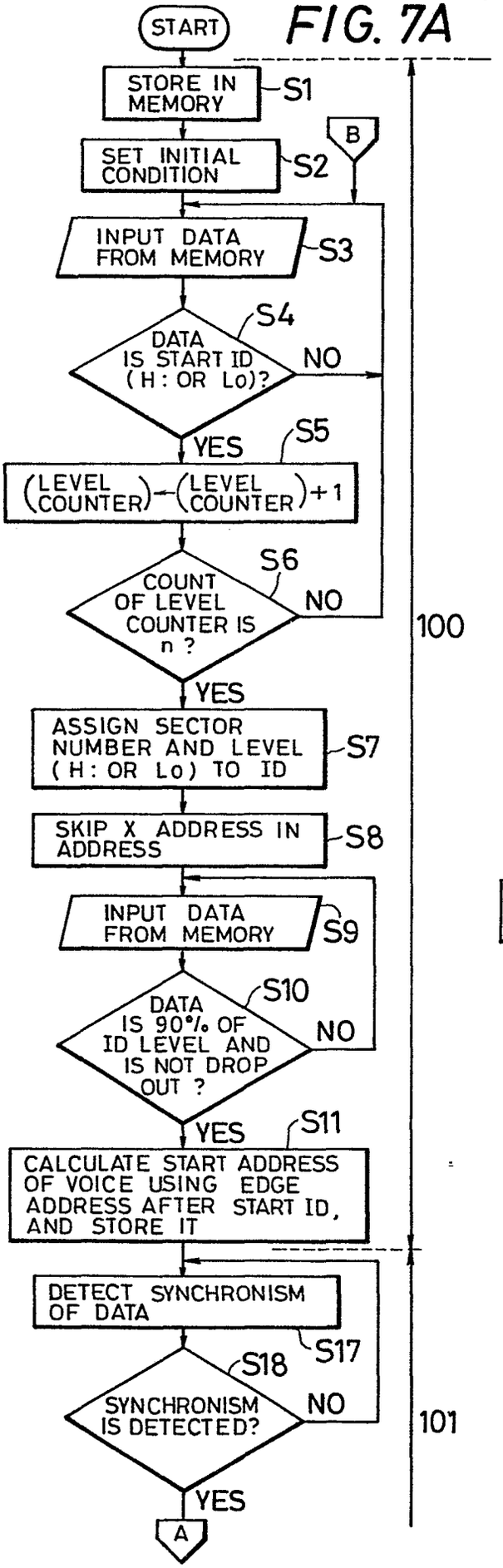

START
STORE IN MEMORY
S1
SET INITIAL CONDITION
S2
B
INPUT DATA FROM MEMORY
S3
DATA IS START ID (H: OR Lo)?
S4
NO
YES
(LEVEL COUNTER) ← (LEVEL COUNTER) + 1
S5
COUNT OF LEVEL COUNTER IS n ?
S6
NO
YES
ASSIGN SECTOR NUMBER AND LEVEL (H: OR Lo) TO ID
S7
SKIP X ADDRESS IN ADDRESS
S8
INPUT DATA FROM MEMORY
S9
DATA IS 90% OF ID LEVEL AND IS NOT DROP OUT ?
S10
NO
YES
CALCULATE START ADDRESS OF VOICE USING EDGE ADDRESS AFTER START ID, AND STORE IT
S11
DETECT SYNCHRONISM OF DATA
S17
SYNCHRONISM IS DETECTED?
S18
NO
YES
A
100
101

FIG. 7B

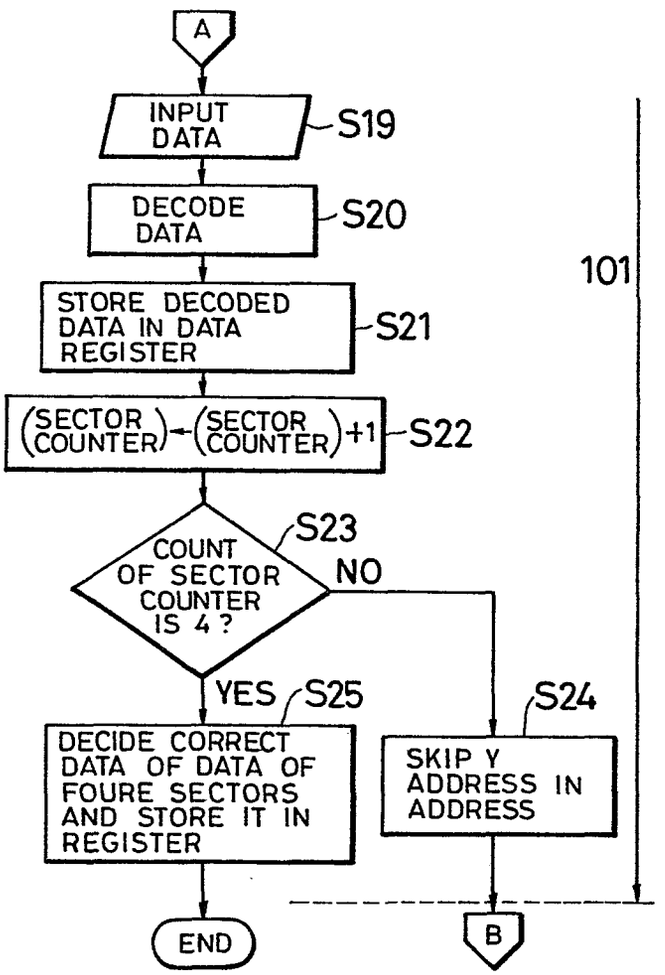

A
INPUT DATA
S19
DECODE DATA
S20
STORE DECODED DATA IN DATA REGISTER
S21
(SECTOR COUNTER) ← (SECTOR COUNTER) + 1
S22
COUNT OF SECTOR COUNTER IS 4 ?
S23
NO
YES
S25
DECIDE CORRECT DATA OF DATA OF FOURE SECTORS AND STORE IT IN REGISTER
END
S24
SKIP Y ADDRESS IN ADDRESS
B
101

FIG. 8A

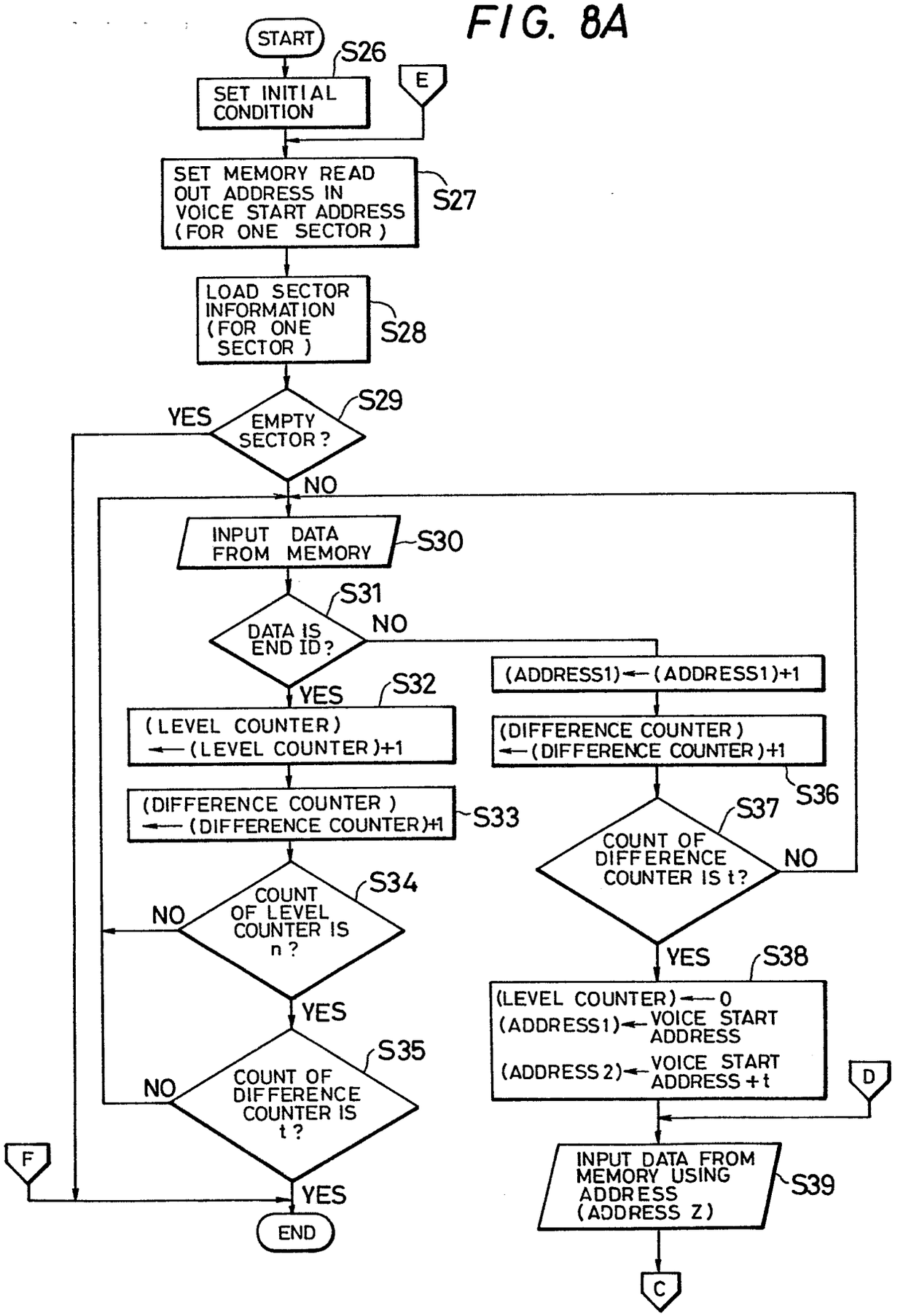

START
S26
SET INITIAL CONDITION
E
SET MEMORY READ OUT ADDRESS IN VOICE START ADDRESS (FOR ONE SECTOR)
S27
LOAD SECTOR INFORMATION (FOR ONE SECTOR)
S28
S29
YES
EMPTY SECTOR?
NO
INPUT DATA FROM MEMORY
S30
S31
DATA IS END ID?
NO
YES
S32
(LEVEL COUNTER) ← (LEVEL COUNTER)+1
(DIFFERENCE COUNTER) ← (DIFFERENCE COUNTER)+1
S33
S34
COUNT OF LEVEL COUNTER IS n?
NO
YES
S35
COUNT OF DIFFERENCE COUNTER IS t?
NO
YES
F
END
(ADDRESS1) ← (ADDRESS1)+1
(DIFFERENCE COUNTER) ← (DIFFERENCE COUNTER)+1
S36
S37
COUNT OF DIFFERENCE COUNTER IS t?
NO
YES
S38
(LEVEL COUNTER) ← 0
(ADDRESS1) ← VOICE START ADDRESS
(ADDRESS 2) ← VOICE START ADDRESS + t
D
INPUT DATA FROM MEMORY USING ADDRESS (ADDRESS Z)
S39
C

FIG. 8B

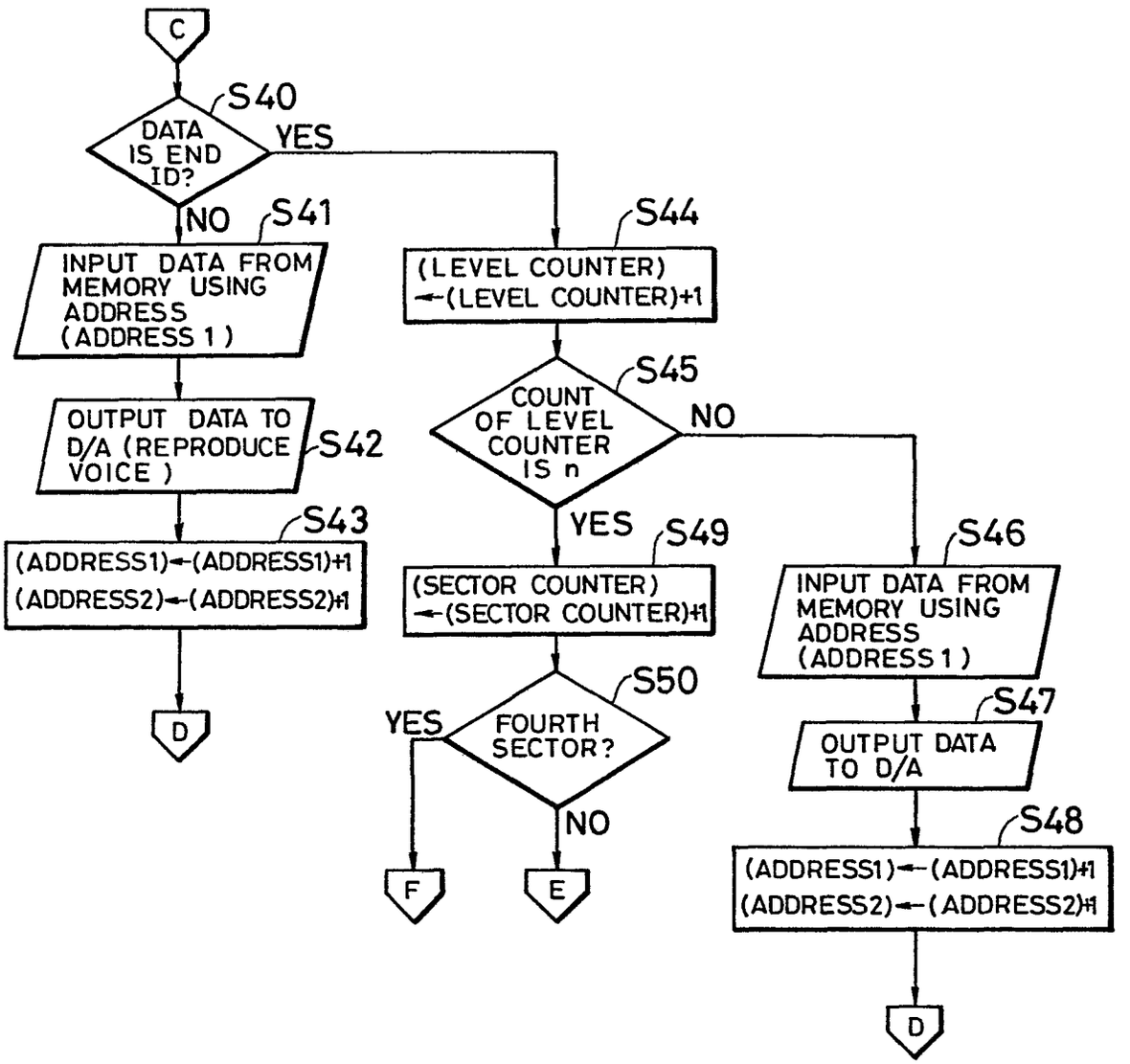

C
S40
DATA IS END ID?
YES
NO
S41
INPUT DATA FROM MEMORY USING ADDRESS (ADDRESS 1)
OUTPUT DATA TO D/A (REPRODUCE VOICE)
S42
S43
(ADDRESS1)←(ADDRESS1)+1
(ADDRESS2)←(ADDRESS2)+1
D
S44
(LEVEL COUNTER) ←(LEVEL COUNTER)+1
S45
COUNT OF LEVEL COUNTER IS n
NO
YES
S49
(SECTOR COUNTER) ←(SECTOR COUNTER)+1
S50
YES
FOURTH SECTOR?
NO
F
E
S46
INPUT DATA FROM MEMORY USING ADDRESS (ADDRESS 1)
S47
OUTPUT DATA TO D/A
S48
(ADDRESS1)←(ADDRESS1)+1
(ADDRESS2)←(ADDRESS2)+1
D

FIG. 9

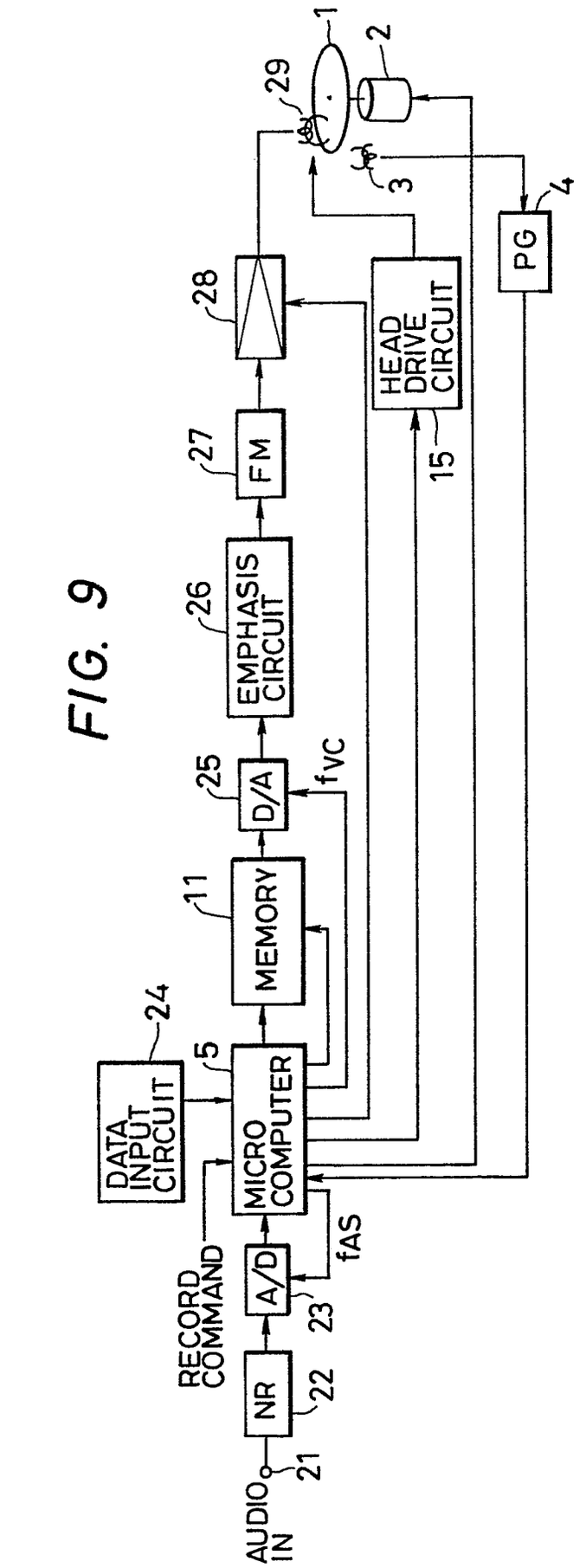
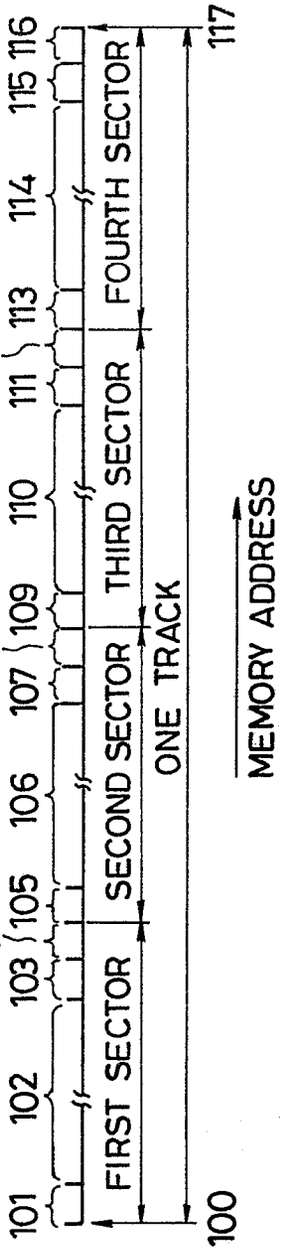

AUDIO IN
21
NR
22
A/D
23
RECORD COMMAND
fAS
MICRO COMPUTER
5
DATA INPUT CIRCUIT
24
MEMORY
11
D/A
25
fVC
EMPHASIS CIRCUIT
26
FM
27
28
HEAD DRIVE CIRCUIT
15
PG
4
3
29
2
1

FIG. 10


101
102
103
104
105
106
107
108
109
110
111
112
113
114
115
116
117
100
FIRST SECTOR
SECOND SECTOR
THIRD SECTOR
FOURTH SECTOR
ONE TRACK
MEMORY ADDRESS